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Efficient Methods for Broadcasting Multi-Slot Messages with Random Access with Capture

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***Abstract*—A new technique for best effort delivery of bursty multi-slot messages is investigated when the messages are broadcast to a group of users with random access with capture. The messages originate at random times from each node within a network and are destined to all other nodes in the network. It is shown how in the absence of any feedback, the proper combination of coding and channel access strategy can improve message reception probability, satisfaction of delay constraints, and throughput in many cases. Furthermore, we show how even in large networks with the optimal use of coding and access strategies, the edge effects of receiver capture due to the finite size of the network significantly impact performance not only at the edges of the network, but also in the center of the network. Finally, we demonstrate how the addition of minimal feedback from a single node can substantially improve the probability of message reception for all nodes in such a network.**

I. Introduction

The increasing demand for transmission of larger files such as images can result in the need to transmit bursty traffic, which can result in a file or message extending beyond a single physical layer frame or time slot, in wireless networks. Such bursty traffic is not well suited to transmission schemes with fixed channel assignments such as TDMA. Random access protocols are good candidates for bursty traffic when there are many bursty users, each with an average low traffic rate. While protocols that reserve the channel such as CSMA/CA or DBTMA [1] are beneficial for point-to-point transmissions, such protocols were not designed to handle delivery of a message to multiple destinations.

ALOHA type protocols could be used to handle bursty broadcast messages, but they generally suffer from collisions and low throughput, as well as delay at high traffic loads. Furthermore there has been relatively little study on the use of ALOHA for multi-slot messages, particularly when they are broadcast. Transmission of point-to-point multi-slot messages over multiple ALOHA channels was accomplished in [2] and [3] with an erasure code covering the multiple slots. In [4] a protocol for transmission of point-to-point multi-slot messages on a single channel was introduced.

In this work, we handle best effort delivery of point-to-multipoint multi-slot messages, and investigate the case of an ALOHA type protocol with capture. These conditions are applicable to Link 16, in which the nearly perfect capture model used here serves as a close approximation to that actually implemented. Furthermore, in Link 16 nodes have knowledge of each others' locations; we leverage this type of knowledge in our efficient feedback protocol at the end of this paper.

The difficulty with multi-slot messages is that for a number of applications if any single packet or slot is lost, the entire message is considered lost, which proves challenging when ALOHA is used owing to the large number of collisions. Therefore, we turn to erasure coding to add redundancy against collisions, as in [4]. Use of coding with random access was also considered in small and moderate sized networks to maximize the throughput in [5] when all nodes are backlogged with packets. In contrast, we consider stochastic arrivals of packets at each node and moderate sized and large networks; we consider *best effort* delivery so that time sensitive newly arriving messages can be delivered without undue delay caused by retransmissions of old messages.

Furthermore, we consider the challenging case of messages that are designed for all nodes in the network. A difficulty with broadcast messages is that with conventional methods the amount of feedback typically grows with network size. Hence, we lastly show how to leverage the capture effect to substantially improve performance with use of only minimal feedback.

If the entire coded message were to be transmitted in consecutive slots, little diversity would be gained by the coding. Therefore, we use packet level erasure coding, along with careful selection of the retransmission probability, to provide favorable tradeoffs between throughput and delay constraints. Our random spreading in time of coded packets introduces a *capture diversity*, that results from different coded packets of a single message receiving very different levels of interference from different interfering nodes. We demonstrate the spatial dependence of performance that results from the capture effect, and we utilize this dependence to further improve performance in the broadcast transmissions through use of a small amount of feedback.

In Section II we described the network and communication model used. Analysis of this model for no spreading of a message in time and for extreme spreading is presented in Section III. Intermediate levels of spreading are investigated with simulation in Section IV. In Section IV-A we discuss the throughput, delay, and loss characteristics derived from simulations when aggregated over all nodes in the network. In

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Section IV-B we consider the spatial variance of performance due to the capture effect and finite network size, and show the substantial benefit of adding a limited amount of feedback to the protocol. A summary is provided in Section V.

II. Network and Protocol Model

We investigate the transmission of multi-slot messages when a slotted ALOHA protocol is used with capture. In order to protect messages against individual packet collisions, we use erasure coding at the packet level and a random spreading of the k packets, as in [4]. Thus each k packet message is transmitted by sending n coded packets. We assume the coding is such that if any k of the coded packets are received by a node, the message is successfully decoded; otherwise, it is not. We further assume a node is able to decode a message as soon as it receives k of the n coded packets, and that all nodes are half-duplex.

Because in this work we consider broadcast transmissions in a network with a total of N nodes and the capture effect, if T nodes each transmit a coded packet simultaneously in a single slot, some of the $N-T$ non-transmitting nodes may capture a coded packet from one transmitting node, while other non-transmitting nodes may instead capture a coded packet from a different transmitter in that time slot.

A line of N equally spaced nodes, as well as a two-dimensional grid of nodes placed on the points of a square lattice, are considered as examples in much of the paper. Since many of the results actually depend only on relative orderings of nodes, rather than on their precise locations, these results also provide insights to many additional node configurations as well. For example, we have considered randomly removing various fractions of nodes from a network that consists of nodes on the 2D grid of a square lattice; for most cases, the performance characteristics were the same as for the fully occupied lattice of nodes. We have considered networks of size ranging from 20 to 400 nodes.

As N is increased from 10 to 20 to 100, significant improvement in throughput is seen at each increase in N, due to the fact that when there are more nodes, the transmit while receive collisions, as well as collisions from equidistant transmitters, become a smaller fraction of the total transmissions. As N is increased beyond a few hundred nodes, little further improvement is seen.

For different sized messages of length k slots, Table I summarizes the optimal coded message size n. In addition, the highest achievable total throughput S is given, along with the corresponding value of total offered traffic G at which this maximum throughput is achieved. When $k=1$, due to the nearly perfect capture, it is beneficial to have the high values of offered traffic. Furthermore, the larger the message size, the lower the throughput due to the finite size block length and the loss of use of all coded packets when fewer than k are received. On the other hand, making the block length too long or using a rateless code would mean that most of the receivers, that is those that have already received k coded packets, would be obtaining no useful throughput during many of the redundant coded packet transmissions.

TABLE I
OPTIMAL CODING AND MAXIMAL THROUGHPUT FOR LARGE NETWORKS WITH INDEPENDENT PACKETS IN EACH MESSAGE.

k	Best n	Highest S	G for Highest S
1	1	.97	5.5
2	3	.54	1.6
3	6	.47	1.4
4	9	.44	1.1

Each message is addressed to all of the other nodes. All nodes have the same probability of message generation, and in any given time slot multiple nodes may be transmitting. Upon generation at a node, the first coded packet of a message is immediately transmitted, if there are no coded packets from preceding messages in the buffer awaiting transmission. If there are coded packets awaiting transmission in the buffer, the new message is placed at the end of the queue. Each of the remaining coded packets, after the first coded packet is transmitted, is transmitted successively in any given time slot with probability Pt. If $Pt=1$, then the protocol is similar to unslotted ALOHA with a packet length of n, and the coded packets offer little diversity since it is likely that multiple coded packets will be lost in any collision. The same is true for large values of Pt less than 1. If Pt is too small, it will take a long time for a node to completely transmit a message.

We consider collisions as the only channel impediment, so that in the absence of concurrent transmission, every node can successfully receive from every other node. We initially consider an ideal capture model in which the node will receive the packet from its nearest transmitting neighbor. In the event that two transmitting nodes are at equal distances from the receiver and that this distance is the shortest distance between the receiver and any transmitting node, the packet will be lost. If a node is transmitting, it is not able to receive any other messages.

A message buffer of four messages per node is employed. The message buffer queues messages that arrive while another message has already begun transmitting. This buffer size was sufficient to avoid all queue drops for the per node offered traffic levels considered here.

III. Analysis for Extremes in Spreading Packets in Time

In this section we consider the two extreme limiting cases of the range of allocation strategies we consider when no bandwidth is allocated to feedback. First, we consider transmitting coded packets spread enough in time so that loss of one coded packet in a message is independent of loss of the other coded packets of the message. Next, we consider transmission of *uncoded* multi-slot messages with no spreading in time.

A. Independent Coded Packets

We explored a range of k, n, and N values for the case when coded packets are independently lost. If all of the coded packets were independently subject to loss from collisions, then the per node average received throughput from

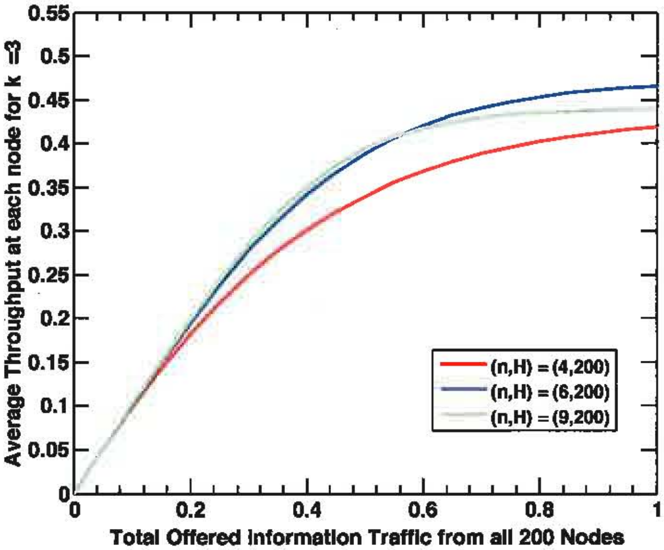

Fig. 1. Throughput in a 200 node network for 3 slot messages that are coded to 4, 6 and 9 slots. The coded packets are subject to independent losses.

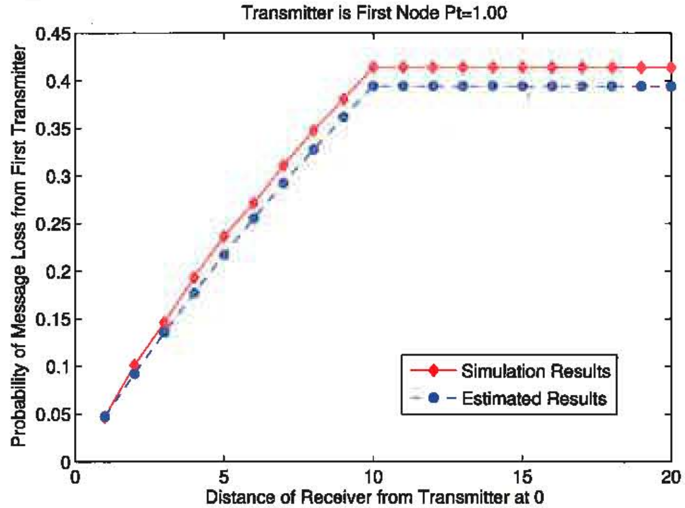

Fig. 2. Probability of loss for uncoded transmissions. Estimated results include collisions between only two nodes at a time, whereas simulated results include collisions between any number of nodes.

all transmitting nodes would be given by

$$\frac{2}{N}\frac{k}{n}g_c\sum_{f=1}^{g-1}\sum_{g=2}^{N}\sum_{j=k}^{n}\binom{n}{j}\exp(-jN_ig_c)\times(1-\exp(-N_ig_c))^{n-j}, \tag{1}$$

where g_c is the per node offered coded traffic, and the number of interfering nodes N_i that each transmission is subject to is

$$N_i = \min(2\times(g-f), N-f). \tag{2}$$

Equation (1) is obtained from considering all pairs of nodes in the line, and assumes that the traffic from N_i nodes is Poisson distributed. This assumption matches the model used here when Pt is small, which is precisely when we expect the losses of coded packets from a single message to be the least correlated. Thus (1) shows the limiting behavior of independent losses on coded packets that we expect for very small Pt.

We first consider $k = 3$ slot messages. We found the choice of $n = 6$ to be the optimal value of n for $k = 3$. We illustrate several choices of n for $k = 3$ in Figure 1 for a 200 node line network with several sample values of n of the many investigated. While transmission of more redundant packets ($n = 9$), is beneficial at low values of traffic, when the traffic becomes higher the additional coded packets cause more collisions, rendering the $n = 6$ more favorable at higher traffic levels. The lower values of n, such as $n = 4$ do not offer enough protection against loss of the whole message if a few coded packets are lost.

B. Transmissions in Bursts

We now consider the case of uncoded packets where each node transmits the entire multi-slot message upon its arrival at the node. In this case, there is no additional energy or time slots allocated to coding, in that every node immediately transmits its entire message. Hence, with the capture effect for a k slot message, the probability of loss at any given node that has N_i interfering nodes closer to it than the transmitting node is the probability that none of the N_i interfering nodes transmit in any of the $2k-1$ time slots that would cause an overlap of the messages:

$$Ploss \approx 1-(1-N_i\times G/N)^{2k-1}. \tag{3}$$

In particular in Figure 2, we plot the message loss probability for $k = 3$, $N = 21$, when we consider the transmitting node at the edge of a line of nodes. We have also plotted simulation results for comparison. Because (3) calculates the loss resulting from collisions involving only two messages, it slightly underestimates the total probability of loss, as shown in Figure 2; the simulated results, include collisions of multiple messages.

IV. Simulations

A. Aggregate Performance

Figure 3 plots the throughput received at a node as a function of the total offered information traffic in the network. This per node received throughput is averaged over all receiving nodes, which is all N nodes. The total offered information traffic is the product of the average number of coded packets transmitted by all N users in a time slot and k/n. The dotted curves represent no coding, whereas the solid curves represent the use of coding.

We next consider delay, and in particular we investigate what fraction of the nodes can satisfy specific delay constraints. When the delay constraint threshold is only five times the message length, in this case a 15 slot delay constraint, we see from Figure 5, which plots the cumulative distribution function (CDF) at 15, that the probability of delay increases as Pt decreases for $Pt \leq .35$. With all of these Pt values, the messages are more likely delayed than lost. For larger values of Pt, further increasing Pt does not necessarily increase the

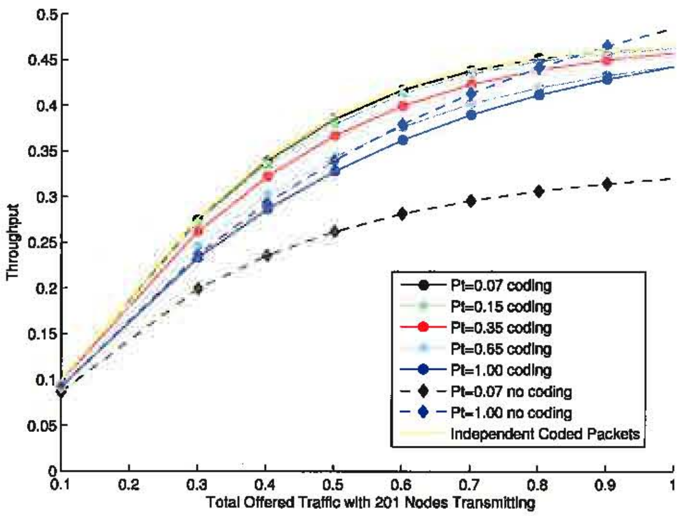

Fig. 3. Throughput received at a node from all other nodes vs. offered information traffic for 201 nodes in a linear network.

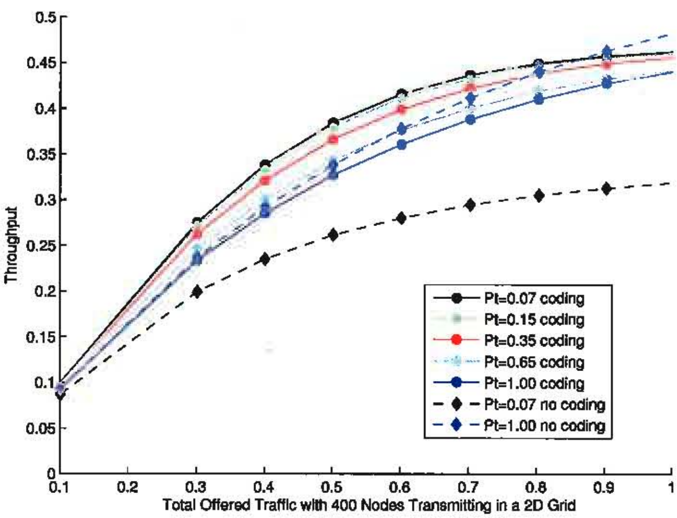

Fig. 4. Throughput received at a node from all other nodes vs. offered information traffic for 400 nodes in a 2-dimensional grid.

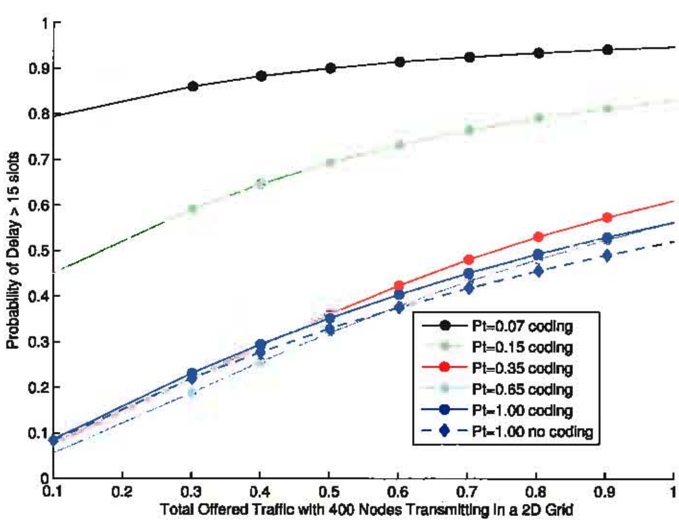

Fig. 5. CDF at 15 slots.

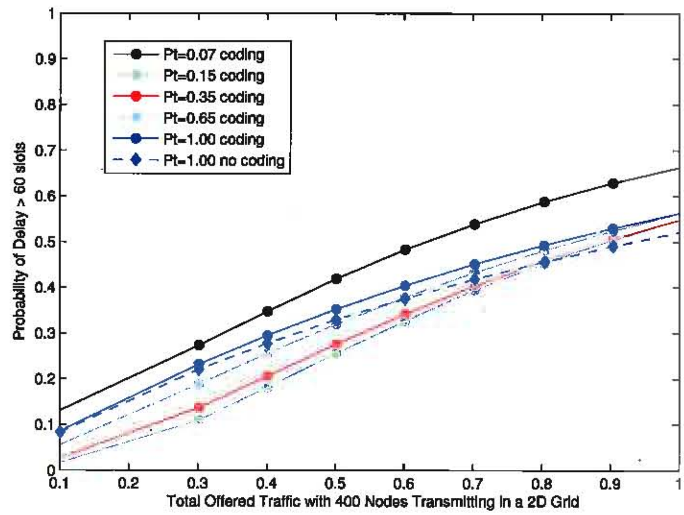

Fig. 6. Cumulative Distribution Function (CDF) at 60 slots.

probability of satisfying this delay constraint threshold; while increasing Pt reduces the transmission time between coded packets, at the large Pt values it also increases the message loss probability, and hence increases the probability of infinite delay.

For larger values of the delay constraint threshold, the effect of message loss on delay is even more pronounced in comparison to the effect of spreading time between coded packets. In Figure 6, the CDF is plotted at a delay threshold of 60 slots, and it is seen to closely resemble the probability of message loss graph, which is plotted in Figure 7. The CDF shows the probability that the message is delayed more than twenty times the length of the message, which is also the approximate length of three TDMA cycles. When looking at delays greater than 60 slots, for all values of Pt explored, except the smallest one of $Pt = .07$, it is shown that they are dominated by the probability of message loss. As shown in Table 1, for $Pt = .15$ and $Pt = 1.0$ for example, the expected number of slots, in the absence of collisions, to send a full message is well below 60. In these cases, a message not being received after 60 slots is most likely lost. In contrast, for Pt =0.07, a larger number of slots is required transmit the message due to greater spreading. In this case, after 60 slots it is still reasonably likely that the transmitter has not finished sending all of the packets, causing the delayed messages to dominate the lost packets.

TABLE II
AVERAGE NUMBER OF SLOTS TO SEND MESSAGE IN ABSENCE OF COLLISIONS

Pt	Slots
1.00	2-5
0.15	14-35
0.07	26-65

As the offered traffic in the system increases, the probability of message loss also increases. Furthermore, larger values of Pt lead to more message loss when coding is used since larger values of Pt mean a higher likelihood of multiple

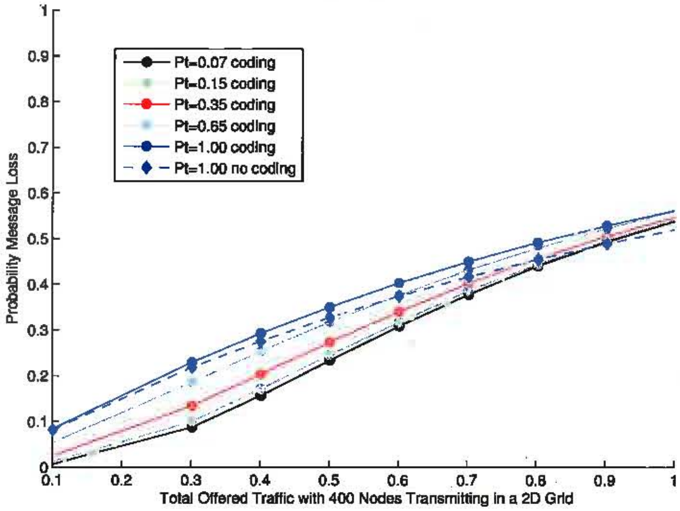

Fig. 7. Probability of message loss vs. offered information traffic.

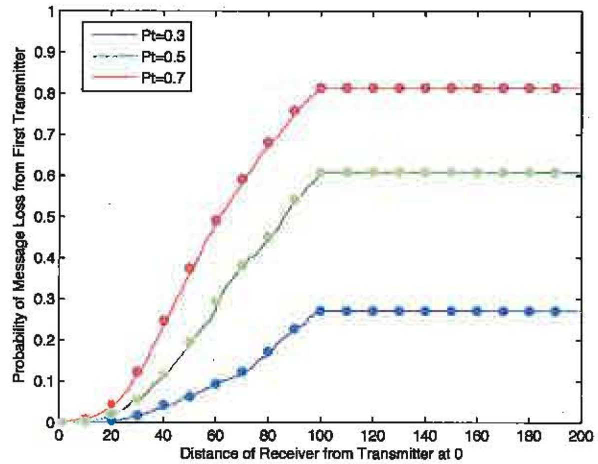

Fig. 8. Message loss vs. distance from the transmitting node for a total offered information traffic of .3.

coded packets from the same two messages colliding as they are more clumped in time, thereby negating the diversity benefit of coding. The dashed lines represent simulations in which n=3, which means that there is no coding used. For uncoded messages, since once a single packet of the message is lost, the entire message is lost, it is more favorable to have multiple slots of messages collide with each other, rather than having additional slots of these already lost messages collide with other additional messages that may still be successfully received. Conversely, when coding is used, message loss is reduced by spreading out the transmitting of the packets, so as to leverage the potential coding diversity, by reducing the Pt value. The use of coding improves message loss until the coded packet traffic is so high that the increase in collisions outweighs the additional redundancy and diversity provided by the coding.

B. Spatial Dependence of Performance

We now explore the spatial dependence of the probability of message loss. The capture effect makes it more likely that nodes close to a transmitter will receive a message than nodes farther away. For the sake of illustration, we consider a 200 node linear network.

Figure 8 shows the probability that a message transmitted by the first node will be lost by each of the receivers for several values of Pt for a total offered information traffic of .3. As shown in the figure, there is a saturation point at node 100 beyond which all future nodes are impacted by the same amount of message loss. Message loss occurs when another node closer to the receiver transmits in the same time slot as node 0, so receivers further away from the transmitter will receive more interference and therefore likelihood of message loss. This plateau point occurs due to edge effects. For example in this case, every node beyond node 10 has an equal number of potential interfering transmitters. For example, for a transmitting node located at $t < N/2$ nodes from the left most transmitter, where N is the total number of nodes in a linear network, the two potential plateau points would be located at transmitters P_1 and P_2:

$$P_1 = \lceil (\frac{N-t}{2}) \rceil + t \quad (4)$$

$$P_2 = \lfloor (t/2) \rfloor \quad (5)$$

In this case N is the number of nodes in the system and P is the plateau point index. This figure also shows that as the offered traffic in the system is increased, the probability of message loss increases. Note: when the middle node is the transmitter there are *two* pivots, defined by each of the equations above, that both must acknowledge message receipt for feedback to be successful.

A similar profile is seen if mean delay is plotted as a function of distance from the transmitting node. More traffic leads to a higher mean delay, as expected. Mean delay increases rapidly with distance from the transmitting node until the plateau point is reached at which point the mean delay stays constant at longer distances due to edge effects.

We now consider packet receptions from transmissions made by the middle node. Figure 9 shows the probability of losing a message that is sent from the central transmitter. This figure shows the expected mirrored loss as the distance from the transmitting node is increased in either direction.

Here we have illustrated the spatial dependence of probability of message loss resulting from the combination of capture and edge effects in a one-dimensional network. The spatial dependence of offered traffic for a fixed level throughput resulting from capture and edge effects for single-slot messages were shown in [6].

We now propose utilizing the spatial dependence of packet loss that is induced by the combination of the capture effect and edge effects. Since in the model used here, with capture and collisions being the only source of packet loss, it is seen from Figure 8 that if any of the nodes between nodes 100 and 200 received the message, then they all received it. Therefore, we implement the following feedback protocol when coding and random spacing in time is used: When the receiver at

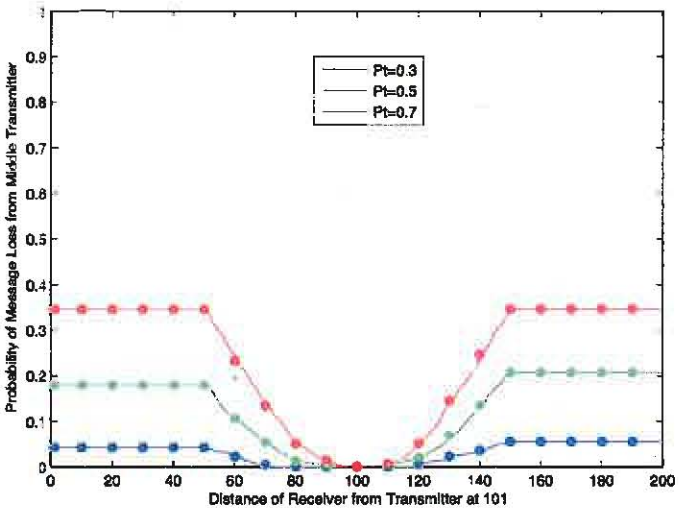

Fig. 9. Percent of Message Loss vs. Distance from Transmitter at total offered information traffic of .3.

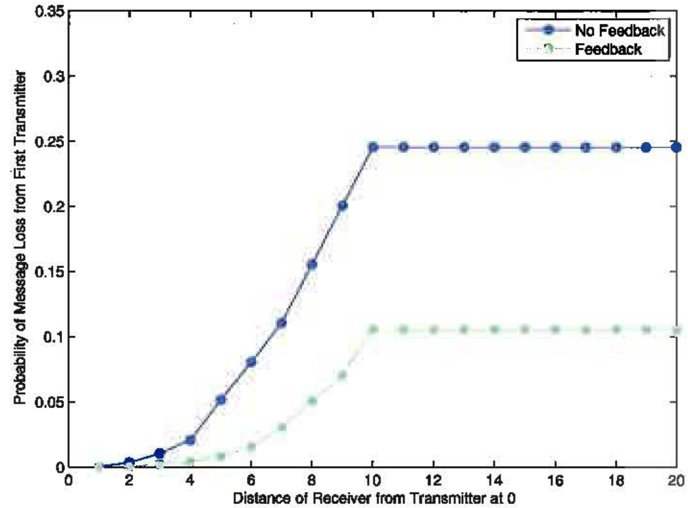

Fig. 10. Feedback to the transmitter at 0.

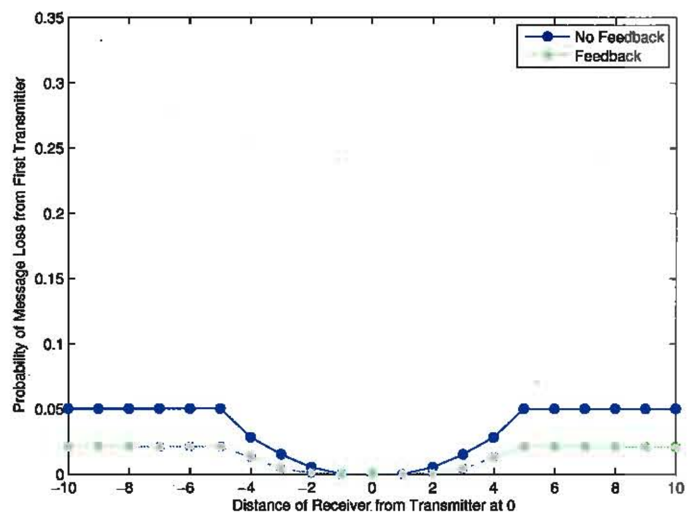

Fig. 11. Feedback to the transmitter at 0.

the designated plateau point for each transmitter receives the message (that is any k coded packets), it sends an acknowledgement to the transmitting node. This acknowledgement signals to the transmitting node whether or not it has finished sending all n packets, and hence whether it can cease sending any additional packets for that message. The plateau point identifies the closest receiving node that will be impacted by every other potential interfering node. When this node has received the message, it can be assumed that every other node has as well. By enabling this feedback, we are able to reduce the overall traffic in the system and therefore reduce collisions and loss, as well as delay, of future messages.

Figure 10 shows the resulting decrease in message loss from this protocol for $N = 21, k = 3, n = 6$, and $Pt = 0.07$, for an offered traffic of .3. The upper curve represents no feedback when all n packets are transmitted, whereas the lower curve shows the performance with the feedback protocol. The difference between the two curves shows the significant reduction in loss provided by the feedback protocol. The savings from the feedback is greatest where the loss is greatest, which is beyond the plateau point. Here, the message loss is reduced by more than a factor of 2.

Similarly, the savings due to the feedback when the center node transmits are shown in Figure 11. Again, for the half of the nodes beyond the plateau point, the message loss is reduced by a factor greater than 2. The absolute value of the decrease in probability of message loss is only a few percent here, whereas it was 15% when the edge node transmitted, because the absolute value of the original message loss probability is smaller when the middle node transmits. Hence, when averaging over all transmitting and receiving nodes, the absolute value of the probability of message loss decreases a few percent when the feedback protocol is used. The savings from using the feedback protocol are greatest for the nodes most distant from the transmitting node.

More generally, if there is partial capture than the plateau points move in towards the transmitter, while the probability of message loss increases at these plateau points. In the limit of no capture, the probability of message loss is greatest, and the nodes adjacent to the transmitter experience the same loss as all the other nodes in the network, and feedback from them would indicate the reception status of all nodes in a pure collision channel. Finally, if in addition to collisions, communication was subject to noise or other interference, then the plateau points could indicate if the remaining nodes definitely need more coded packets. They could also indicate with a probability depending on noise, that the remaining nodes have all received the entire message. Such a feedback mechanism could be extended to reliable communications as well. These topics are areas for future investigation.

Finally, we note that while some communication systems, such as Link 16, exhibit nearly perfect capture, others may have an imperfect capture. Thus in Figure 12, we show the effect of weak capture, as compared to that of nearly perfect capture. In this weak capture model, a receiver can capture the signal from transmitter T, if no other transmitter is within

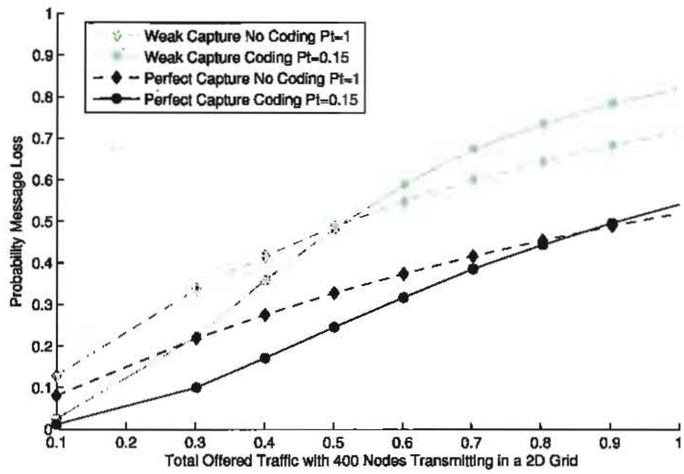

Fig. 12. Overall comparison for 2-dimensional grid.

twice the distance of the receiver that T is.

V. Summary

The use of erasure coding, together with the random spreading in time of individual coded packets of a message to create "capture diversity", enables significant throughput gains in multi-slot messages transmitted with random access by multiple sources. It is shown how the amount of spreading can be selected to satisfy delay constraints, and that when coding is not used, spreading should not be used. Finally, it is shown that edge effects and capture result in an increase in message loss with distance from the transmitting node, until a saturation point is reached. Finally, we propose to use this phenomenon and demonstrate how it can enable consolidation of feedback and elimination of future unnecessary transmissions.

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